

Player

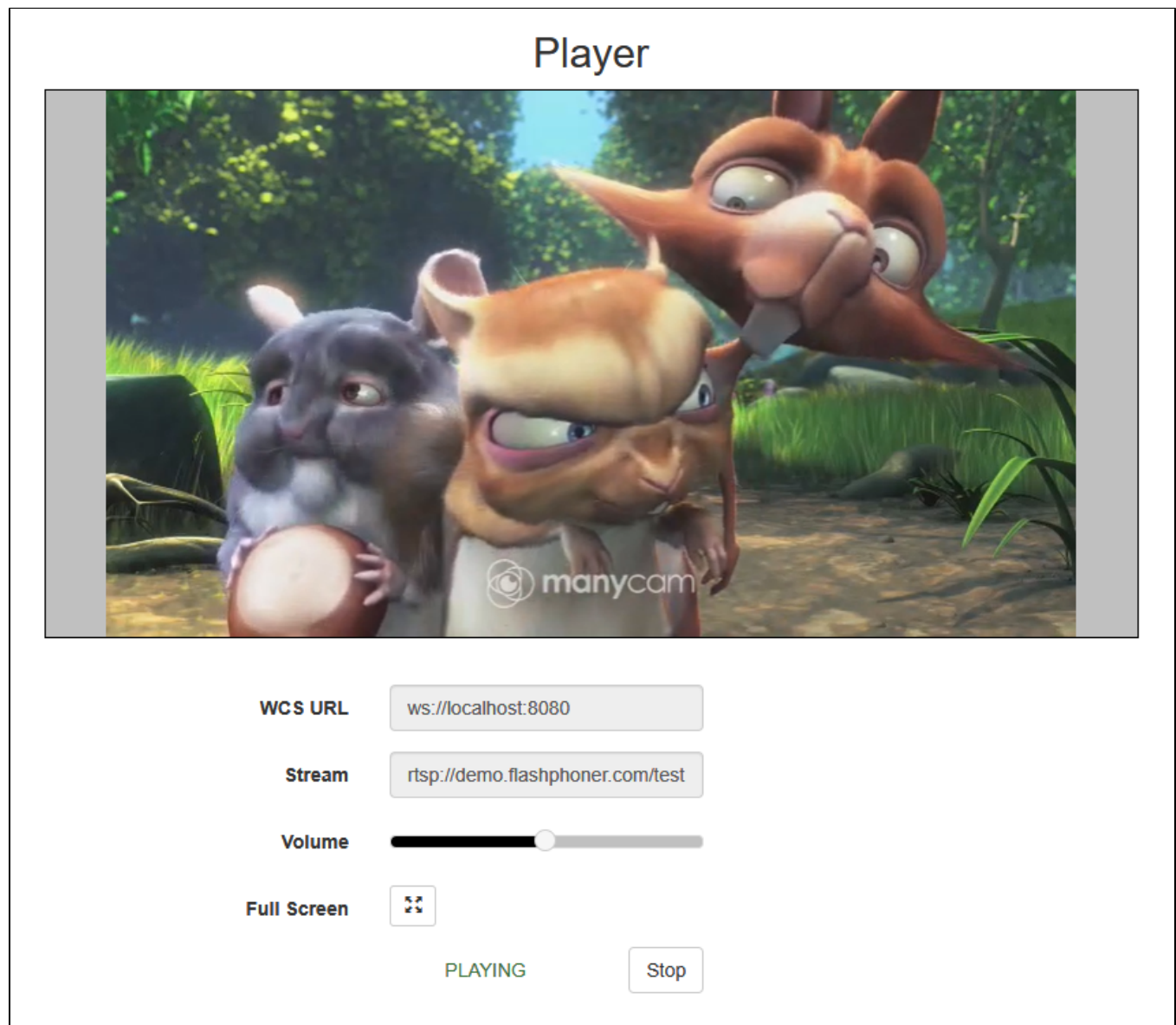
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Player example

This player can be used to play any type of stream on Web Call Server

- RTSP
- WebRTC
- RTMP

On the screenshot below an RTSP stream is being playing in the web player.



Code of the example

The path to the source code of the example on WCS server is:

/usr/local/FlashphonerWebCallServer/client2/examples/demo/streaming/player

player.css - file with styles
player.html - page of the player
player.js - script providing functionality for the player

This example can be tested using the following address:

<https://host:8888/client2/examples/demo/streaming/player/player.html>

Here host is the address of the WCS server.

Analyzing the code

To analyze the code, let's take the version of file player.js with hash 7fff01f, which is available [here](#) and can be downloaded with corresponding build [2.0.219](#).

1. Initialization of the API

Flashphoner.init() [code](#)

```
Flashphoner.init({
  receiverLocation: '../dependencies/websocket-player/wsReceiver2.js',
  decoderLocation: '../dependencies/websocket-player/video-worker2.js',
  preferredMediaProvider: mediaProvider
});
```

2. Connection to server

Flashphoner.createSession() [code](#)

```
Flashphoner.createSession({urlServer: url}).on(SESSION_STATUS.ESTABLISHED, function(session){
  ...
}).on(SESSION_STATUS.DISCONNECTED, function(){
  ...
}).on(SESSION_STATUS.FAILED, function(){
  ...
});
```

3. Receiving the event confirming successful connection

SESSION_STATUS.ESTABLISHED [code](#)

```
Flashphoner.createSession({urlServer: url}).on(SESSION_STATUS.ESTABLISHED, function(session){
  setStatus(session.status());
  //session connected, start playback
  playStream(session);
}).on(SESSION_STATUS.DISCONNECTED, function(){
  ...
}).on(SESSION_STATUS.FAILED, function(){
  ...
});
```

4. Playback of video stream

Session.createStream(), Stream.play() [code](#)

```
stream = session.createStream(options).on(STREAM_STATUS.PENDING, function (stream) {
  ...
});
stream.play();
```

When stream is created, the following parameters are passed

- streamName - name of the stream (rtsp:// address in this case)
- remoteVideo - div element, in which the video will be displayed
- picture resolution to transcode
- unmutePlayOnStart: false - disables automatic audio unmuting for autoplay to conform browsers requirements

[code](#)

```
var options = {
  name: streamName,
  display: remoteVideo,
  flashShowFullScreenButton: true
};
if (Flashphoner.getMediaProviders()[0] === "MSE" && mseCutByIFrameOnly) {
  options.mediaConnectionConstraints = {
    cutByIFrameOnly: mseCutByIFrameOnly
  }
}
if (resolution_for_wsplayer) {
  options.playWidth = resolution_for_wsplayer.playWidth;
  options.playHeight = resolution_for_wsplayer.playHeight;
} else if (resolution) {
  options.playWidth = resolution.split("x")[0];
  options.playHeight = resolution.split("x")[1];
}
if (autoplay) {
  options.unmutePlayOnStart = false;
}
```

5. Receiving the event confirming successful stream playback

STREAM_STATUS.PLAYING [code](#)

```
stream = session.createStream(options).on(STREAM_STATUS.PENDING, function (stream) {
  ...
}).on(STREAM_STATUS.PLAYING, function (stream) {
  $("#preloader").hide();
  setStatus(stream.status());
  onStarted(stream);
}).on(STREAM_STATUS.STOPPED, function() {
  ...
}).on(STREAM_STATUS.FAILED, function() {
  ...
});
stream.play();
```

6. Handling the event about channel bandwidth

STREAM_EVENT, STREAM_EVENT_TYPE.NOT_ENOUGH_BANDWIDTH [code](#)

```
}).on(STREAM_EVENT, function(streamEvent){
  if (STREAM_EVENT_TYPE.NOT_ENOUGH_BANDWIDTH === streamEvent.type) {
    var info = streamEvent.payload.info.split("/");
    var remoteBitrate = info[0];
    var networkBandwidth = info[1];
    console.log("Not enough bandwidth, consider using lower video resolution or bitrate. Bandwidth " +
      (Math.round(networkBandwidth / 1000)) + " bitrate " + (Math.round(remoteBitrate / 1000)));
  } else if (STREAM_EVENT_TYPE.RESIZE === streamEvent.type) {
    ...
  }
});
```

7. Handling the event about changing stream picture size

STREAM_EVENT, STREAM_EVENT_TYPE.RESIZE [code](#)

```

    }).on(STREAM_EVENT, function(streamEvent){
        ...
    } else if (STREAM_EVENT_TYPE.RESIZE === streamEvent.type) {
        console.log("New video size: " + streamEvent.payload.StreamerVideoWidth + "x" + streamEvent.payload.
streamerVideoHeight);
    }
    });

```

8. Playback stop

Stream.stop() [code](#)

```

function onStart(stream) {
    $("#playBtn").text("Stop").off('click').click(function(){
        $(this).prop('disabled', true);
        stream.stop();
    }).prop('disabled', false);
    ...
}

```

9. Receiving the event confirming successful stream playback stop

STREAM_STATUS.STOPPED [code](#)

```

stream = session.createStream(options).on(STREAM_STATUS.PENDING, function (stream) {
    ...
}).on(STREAM_STATUS.STOPPED, function () {
    $("#preloader").hide();
    setStatus(STREAM_STATUS.STOPPED);
    onStopped();
}).on(STREAM_STATUS.FAILED, function() {
    ...
}).play();

```

10. Playback volume control

Stream.unmuteRemoteAudio(), Stream.setVolume() [code](#)

```

$("#volumeControl").slider({
    range: "min",
    min: 0,
    max: 100,
    value: currentVolumeValue,
    step: 10,
    animate: true,
    slide: function(event, ui) {
        //WCS-2375. fixed autoplay in ios safari
        if (stream.isRemoteAudioMuted()) {
            stream.unmuteRemoteAudio();
        }
        currentVolumeValue = ui.value;
        stream.setVolume(currentVolumeValue);
    }
}).slider("disable");

```

11. Playback automatic start on page load

[code](#)

```

if (autoplay) {
    // We can start autoplay with muted audio only, so set volume slider to 0 #WCS-2425
    $("#volumeControl").slider('value', 0);
    $("#playBtn").click();
}

```

